COMMUNICATION MANAGEMENT SYSTEM WITH LINE STATUS NOTIFICATION FOR KEY SWITCH EMULATION

Cross Reference to Related Applications

5   The present application is a continuation in part of United States Patent Application 09/961,532 titled Teledata Space and Docking Station with Modular and Integrated Display filed on September 24, 2001, and is a continuation in part of United States Patent Application 10/000,543 filed on October 23, 2001, titled Modular Multi-Media Communication Management System, and is a continuation in 10   part of United States Patent Application 10/079,128 filed on February 20, 2002 titled Multi-Media Communication Management System with Multicast Messaging Capabilities, and is a continuation in part of United States Patent Application 10/624,815 filed on July 22, 2003 entitled Communication Management System with Line Status Notification for Key Switch Emulation, and the contents of such 15   patent applications is incorporated herein.

Technical Field

The present invention relates generally to managing multi-media communications, and more particularly to a packet-switched communication system 20   with key switch system emulation.

Background of the Invention

In an office environment, desk top telephone service, voice mail service, and fax service is typically provided by a private telephone communication system. 25   Private telephone systems comprises key switch systems, private exchange (PBX) systems, and more recently, voice-over-internet-protocol (VOIP) systems.

A key switch system consists of central key switch unit and a plurality of desk top telephones and fax machines. Each telephone is coupled to the key switch unit by a subscriber loop running from the telephone to the key switch unit. 30   Communication between each desk top telephone and the key switch unit typically utilizes plain old telephone service (POTS) signaling.

The key switch unit may be coupled to a voice mail server by a proprietary interface. The key switch unit is further coupled to the public switched telephone network (PSTN) using trunk lines that are connected to a central office switch that is typically managed by the local telephone service provider.

5   Each telephone includes a plurality of buttons (and illuminated indicators), each being associated with one extension or one of the trunk lines to the central office. When an extension or a trunk line is in use, the indicator associated with that extension will be illuminated on all of the other telephones such that the operators can view which extensions and trunks are in use, and which are

10   available. When an operator desires to telephone another extension, he or she will activate the button associated with the other extension. When an operator desires to telephone a subscriber loop of the PSTN, the operator first activates a button associated with an outside line (which will cause the key switch unit to couple an outside line to the operator's extension giving the operator's extension a PSTN dial

15   tone) and then dial the number.

A PBX system comprises a switching network, a plurality of desk top telephones, and a voice mail server. The voice mail server is typically coupled to the switching network using a proprietary interface. Each desk top telephone and fax machine is coupled to the switching network by an extension line that consists

20   of twisted pair conductors that are terminated by a telephone jack in the office. Communication between the desk top telephone and the switching network over each extension line utilizes either proprietary digital signaling or plain old telephone service (POTS) signaling. The switching network is further coupled to the public switched telephone network (PSTN) using trunk lines that are connected to a

25   central office switch that is typically managed by the local telephone service provider. The switching network controls calls between extensions and between an extension and a remote destination via a trunk line coupled to the PSTN.

PBX systems do not typically have a button associated with each extension on each telephone. Instead, operators dial each other's extension numbers for

30   placing internal calls and typically dial "9" to get an outside line for placing calls over the PSTN.

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VOIP systems utilize a central gateway and a plurality of VOIP telephone stations. Each station is coupled to a packet switched network (such as an Ethernet network) that interconnects each station with the central gateway. The central gateway and each telephone station typically is assigned an IP address and communications between two extensions and between an extension and the gateway typically is over UDP/IP channels. Because of the wide spread acceptance of PBX systems, VOIP systems typically are configured such that operators dial each other's extension numbers for placing internal calls and typically dial "9" to indicate a call to be placed over the PSTN.

It is generally recognized the dialing of extension numbers and dialing "9" for an outside line has several advantages over key systems, particularly in larger environment where there are many extensions. However, in a small office environment, key switch systems have some advantages over PBX systems.

First, each operator can quickly determine whether another operator is on the telephone without having to first attempt to place a call and receive a busy signal (or roll to voice mail) if the operator is on the telephone. Secondly, each operator can quickly determine whether an outside line is available without having to first try to place a call and receive a busy signal if an outside line is not available. Thirdly, an incoming call on one of the trunk lines can typically be answered by any extension simply by activating the button associated with the ringing trunk line.

What is needed is a communication management system that provides a typical key switch user experience while providing the flexibility of VOIP networking.

Summary of the Invention

A first aspect of the present invention is to provide a real time communication device for operation with a packet switched network interconnecting the real time communication device with a control unit and a plurality of remote real time communication devices. The real time communication device comprises a network interface for communicating over the packet switched network.

The real time communication device further comprises means for establishing a logical channel to support a media session over the packet switched

network with an endpoint for the exchange of real time streaming media with the endpoint during a media session. The endpoint may be selected from the group of endpoints consisting of the control unit and a one of the plurality of remote real time communication devices.

5   The real time communication device may further comprise: i) means for receiving microphone input and generating compressed digital audio frames representative thereof for transmission to the endpoint during the media session, and ii) means for receiving compressed digital audio frames from the endpoint and driving a speaker to output audio in response thereto;

10   The real time communication device also comprises means for sending a multicast status message on the packet switched network addressed to a multicast group. The multicast status message may announce a state of the real time communication device. The state is a state selected from a group of states consisting of a first state wherein the real time communication device is

15   participating in a media session and a second state wherein the real time communication device is not participating in a media session. The multicast group may comprise any of the remote real time communication devices that have joined the multicast group.

        Further, the real time communication device may comprise means for

20   obtaining a first IP multicast address - either hard coded, semi-permanently stored in a non volatile memory, or obtained from the control unit. And, the means for sending a multicast status message may comprise   sending   the   multicast message to the first IP multicast address. The multicast status message may include identification of the real time communication device in addition to an

25   indication of the state of the real time communication device.

        Further yet, the real time communication device may comprise a user interface that includes an indication of the state of each of the remote real time communication devices and means for receiving a plurality of multicast status messages. Each multicast status message may be sent by one of the remote real

30   time communication devices and comprise identification of the remote real time communication device that sent the multicast status message and identifying the

state of the remote real time communication device that sent the multicast status message. T he real time c ommunication d evice may f urther comprise means for driving the indication of the user i nterface to show the state of each remote r eal time communication device and updating the state of one of the remote real time

5   communication devices in response to receiving a multicast status message sent by the one of the remote real time communication devices.

In one embodiment, the multicast status messages sent by each remote real time communication device may be sent to the first IP multicast group which includes all of the real time communication devices.

10   In a second embodiment, the real time communication devices may be divided into a plurality of groups, each group being assigned a unique multicast address. In this second embodiment, the multicast status messages sent by each remote real time communication device may be sent to the unique multicast address associated with the group to which the device is assigned.

15   In a third embodiment, each resource may be assigned its own unique multicast group. In this third embodiment, multicast status messages sent by each of the remote real time communication devices may each be sent to the address assigned to the resource.

In a first variation, the means for sending the multicast status message

20   comprises means for sending the multicast status message in response to a change in state of the real time communication device between the first state and the second state.

In a second variation, the means for sending a multicast status message comprises means for sending the multicast status message in response to

25   receiving a status refresh message on the multicast channel.

In a third variation, the means for sending a multicast status message comprises means for sending the multicast status message in response to passage of a time duration during following sending of a previous multicast status message.

30   For a better understanding of the present invention, together with other and further aspects thereof, reference is made to the following description, taken in

conjunction with the accompanying drawings. The scope of the invention is set forth in the appended clams.

Brief Description of the Drawings

5   Figure 1 is a block diagram representing a multi-media communication management system in accordance with one embodiment of the present invention;

Figure 2 is a block diagram representing an exemplary subscriber telephony station;

Figure 3 is a block diagram representing an exemplary wireless telephony

10   device;

Figure 4 is a block diagram representing an exemplary personal data device;

Figure 5 is a block diagram representing an exemplary control unit;

Figure 6 is an exemplary routing table;

Figure 7 is a flow chart representing one aspect of exemplary operation of a

15   a PSTN gateway;

Figure 8 is a flow chart representing exemplary operation of a session initiation proxy server;

Figure 9 is a block diagram representing an exemplary conference server;

Figure 10 is a flow chart representing exemplary operation of a conference

20   server application;

Figure 11a represents an exemplary conference initiation document;

Figure 11b represents an exemplary session status document;

Figure 12a represents an exemplary single frame video display document;

Figure 12b represents an exemplary multi frame video display document;

25   Figure 13 represents an exemplary session status table;

Figures 14a, 14b, and 14c are each flow charts that represent operation of a web server management application in accordance with one embodiment of the present invention;

Figure 15 represents an exemplary main menu display document;

30   Figures 16a and 16b are each flow charts that represent exemplary operation of an email module;

Figure 17 is a flow chart representing exemplary operation of a multicast application in accordance with one embodiment of the present invention;

Figure 18 represents an exemplary paging initiation document;

Figure 19 is a block diagram representing an exemplary status application in accordance with the present invention;

Figure 20 is a flow chart representing operation of a multicast host module of the status application in accordance with an embodiment of the present invention;

Figure 21 is a flow chart representing operation of a multicast client module of the status application in accordance with an embodiment of the present invention;

Figure 22 represents an exemplary resource status table of the status application in accordance with the present invention;

Figure 23 represents an exemplary multicast status message in accordance with the present invention;

Figure 24 is a block diagram representing an exemplary master status application in accordance with the present invention; and

Figure 25 is a flow chart representing exemplary operation of a indicator control module of the status application of the present invention.

Description of the Preferred Embodiments

The present invention is now described in detail with reference to the drawings. In the drawings, each element with a reference number is similar to other elements with the same reference number independent of any letter designation following the reference number.

It should also be appreciated that many of the elements discussed in this specification may be implemented in hardware circuit(s), a processor executing software code, or a combination of a hardware circuit and a processor executing code. As such, the term circuit or module as used throughout this specification is intended to encompass a hardware circuit (whether discrete elements or an integrated circuit block), a processor executing code, or a combination of a hardware circuit and a processor executing code, or other combinations of the

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above known to those skilled in the art.

Figure 1, is a diagram representing an architecture of a multi-media communication management system 10 of the present invention. The system 10 is coupled to both of a multi-media service provider network 18 and the public

5 switched telephone network (PSTN) 42.

The service provider network 18 may be IP compliant and utilize a combination of one or more of co-axial cable, fiber optic cable, T1 lines, and wireless RF channels as its physical communication medium. The service provider network 18 may couple to the Internet 30 through appropriate gateways and/or

10 routers to enable system 10 to communicate with IP compliant devices coupled to the service provider network 18 or the Internet 30 (collectively, remote IP devices).

The system 10 includes a control unit 12, a plurality of network devices 20, a plurality of personal data devices 21, and a wireless local area network 22 that interconnects the network devices 20 to each other and to the control unit 12.

15 The wireless local area network 22 may be an IP compliant packet switched network and utilize a combination of a wired backbone network 23 (such as an Ethernet Network) and micro-cellular RF cells as its physical communication medium. Each micro-cellular RF cell may be an 802.11 compliant wireless cell controlled by an access point 19 that is uplink coupled to the backbone network 23

20 or wirelessly uplink coupled to another access point 19.

The control unit 12 includes a LAN interface circuit 29 for coupling to the local area network 22 and enabling communication with network devices 20, a PSTN interface circuit 13 for coupling to the PSTN 42 (e.g coupling to multiple telephone lines from a telephone service provider central office) and enabling

25 communication with remote PSTN devices coupled to the PSTN 42, and a modular service provider interface 16 for coupling to the service provider network 18 and enabling communication with remote TCP/IP compliant devices.

The network devices 20 may include data devices 17 such as traditional computer systems 32, network printers 46, various network appliances 34 and real

30 time communication devices 15 such as subscriber telephony stations 24 and wirelessly telephony devices 26.

Each personal data device (PDA) 21 may be similar to a commercially available device known as a Personal Data Assistant (PDA), or a commercially available cellular or PCS telephone with PDA capabilities, and may include a point-to-point communication system 62 for communication with a corresponding point-to-point communication system 62 within a station 24.

Subscriber Stations

Each telephony station 24 is configured to provide a subscriber interface that includes aspects of a user interface of a key switch private telephone system. More specifically, each telephony station 24 comprises a plurality of LED lights or LCD display indicators 139 each associated with one of the other real time communication devices 15 of the system 10 or one of the plurality telephone lines 33 coupling the control unit 12 to the PSTN 42 (e.g. an outside line). The indicator 139 indicates the "on-hook" or "off-hook" status of such other real time communication device 15 or the PSTN outside line 33. The display state of each indicator 139 is driven by the status application 136 (Figure 19 discussed herein).

Associated with each indicator 139 is a single one of the plurality of selection buttons 60c that may be activated by the user of station 24 to initiate a real time communication session to the real time communication device 15 associated with the indicator 139 or to a particular logical port number of the PSTN gateway 131 (Figure 5) for establishing a real time communication channel over an outside line 33 associated with the indicator 139.

Also included in each telephone station 24 are a speaker 100 and a microphone 102 for providing a "speaker phone" dialog interface, a handset 98 for providing a traditional handset dialog interface, a display screen or touch panel display screen 59, a plurality of buttons 60a arranged as a traditional telephone keypad, and a plurality of buttons 60b arranged for menu navigation and selection from the display 59.

Referring to the block diagram of Figure 2 in conjunction with Figure 1, exemplary structure of the telephony station 24 is shown. The station 24 may include a controller 112 and a plurality of peripheral circuits coupled to the

controller 12 via a local bus 116. The peripheral circuits may include a wireless module 94, an network interface circuit (NIC) 125, a power management controller 120, compression/encryption hardware (CODEC) 122, a key switch controller 126, a display/touch panel controller 128, a camera controller 72, a indicator circuit 138,

5 a PSTN converter 146, a dialog system 130, and at least one point-to-point wireless communication module 62.

The dialog system 130 may be coupled between the bus 116 and the handset 98 (and the speaker 100 and microphone 102) and may include applicable circuits for: i) driving the speaker 100 (or the speaker in the handset 98)

10 in accordance with digital audio received from the CODEC 122 and ii) detecting input from the microphone 102 (or the microphone in the handset 98) and generating digital audio for presentation to the CODEC 122.

The power management controller 120 selectively receives input power from a battery 70, external power source 134, and/or the backbone network 23 via the

15 network interface circuit 125. The power management controller 120 includes appropriate circuits for converting the input power to appropriate operating power required by each component of the station 24. Additionally, the power management controller 120 includes appropriate circuits for managing charging of the battery 70 when power is available from the external power source 134 or the

20 backbone network 23.

The display/touch panel controller 128 couples to the bus 116, operates under control of applicable drivers operated by the controller 112, and enables the display 59a (or the touch panel display 59b) to provide information to the subscriber (and receive subscriber input through the touch panel display 59b). In the

25 exemplary embodiment, the display/touch panel controllers 128 may include a separate display control circuit compatible with the resolution and color depth of the display 59a or 59b and a touch panel control circuit for detecting subscriber contact with the touch panel 59b.

The camera controller 72 couples to the bus 116, operates under control of

30 applicable drivers operated by the controller 112, and generates digital still image or motion video signals for presentation to the CODEC 112 for transmission to

another endpoint during a VOIP media session and/or for presentation to another applicable application operated by the controller 112 for display on the display 59.

The key switch controller 126 couples to the bus 116, operates under control of applicable drivers operated by the controller 112, and enables the controller 112 to receive subscriber input through the buttons 60.

The wireless point-to-point communication module 62 couples to the bus 116 (either directly or through an interface circuit such as a serial communication controller), operates under control of applicable drivers operated by the controller 112, and enables synchronization of data between the station 24 and the PDA 21 and enables operation of the keyboard 28 (the PDA 21 includes a corresponding wireless point-to-point communication module 62).

Exemplary point-to-point communication modules 62 include known modules that couple to a bus 116 through a serial communication circuit and utilize the IRDA standard or the Blue-Tooth standard for wireless data transfer.

The PSTN converter 146 couples to the bus 116, operates under control of applicable drivers operated by the controller 112, and provides an FXS port 82a for coupling to a PSTN line and/or an FXO port 82b for supporting operation of a traditional telephone or fax machine.

The CODEC 122 couples to the bus 116, operates under control of applicable drivers and a packet voice video application 113 operated by the controller 112. The CODEC 122 includes hardware circuits with adequate operating speed to: i) compress (and optionally encrypt) digital audio provided by the dialog system 130 and digital video provided by the camera controller 72 into sequences of RTP frames for sending to another VOIP endpoint during a media session; and ii) sequence, decompress (and optionally decrypt) RTP frames provided by the other VOIP endpoint into digital audio for presentation to the dialog system 130 and into digital video for display on the display 59.

The indicator circuit 138 couples to the bus 116 and operates under control of applicable drivers and a status application 136 operated by the controller 112. The indicator circuit 138 includes circuits for illuminating each indicator 139 in accordance with instructions provided by the status application 136.

Both the wireless module 94 and the NIC 125 couple to the bus 116 (either directly or through an interface circuit such as a PCMCIA controller), operate under control of applicable drivers operated by the controller 112, and enable the station 24 to communicate with other devices over the network 23.

5   The wireless module 94 provides coupling via a wireless link to an access point 19 while the network interface circuit 125 provides coupling by a direct connection to the backbone network 23 via an uplink port 84a. Exemplary wireless modules 9 4 may i nclude c ommercially a vailable wireless n etwork i nterface cards that wirelessly communicate with an access point 19 utilizing an IEEE 802.11

10   protocol.   Exemplary network interface circuits 125 may include commercially available network interface cards that communicate with a network hub or switch using an Ethernet protocol.

The NIC 125 may include switch circuitry 124a that enables the station 24 to operate a s a network s witch between t he u plink p ort 84a and m ultiple d own l ink

15   ports 84b.   Similarly, the wireless module 94 may include access point circuitry 124b which enables the wireless module 94 to operate as a wireless access point managing wireless communication within its own micro-cell and operate as a switch between uplink communications with an access point 19 (on the channel established by the access point 19) and downlink communications within the micro-

20   cell on a channel established by and controlled by the wireless module 94.

Further, both the wireless module 94 and the NIC 125 may include applicable circuits for communicating frames with each other such that: i) uplink communication utilizes wireless module 94 while downlink communication utilizes the NIC 125; or ii) uplink communication utilizes the NIC 125 while downlink

25   communication utilizes the wireless module 94.

The controller 112 may operate the above discussed drivers, the packet audio/video communication client 113, a system client application 115, a subscriber device application 119, the status application 136, and a multicast application 117.

The system client application 115 enables the station 24 to function as a

30   client to web server applications. An exemplary client application 115 may be a known web browser that provides for: a) initiating an IP connection or channel to a

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web server application; b) g enerating a n image o n t he display 59 i n accordance with a display document or display content and a style sheet received from a web server; c) output of digital audio representing an audio stream file to the dialog system 130; and d) execution of processing steps in accordance with script instructions received from a web server. Such processing steps may include providing messages or posts to the web server indicating subscriber actions (such as keyboard entry, keypad entry, or touch panel entry) and may also include providing an instruction to the communication client 113 to set up a media session in accordance with an identifier provided by the client application 115.

The multi cast module 117 may be a commercially available multicast client compatible with the IP Multicast standard and provides for the station 24 to receive invitations to multicast groups, join multicast groups, and couple received multicast media to the dialog system 130 or the display 59 for output.

The subscriber device application 119 provides application layer coupling to a corresponding a pplication 51 in each PDA 2 1 (Figure 4) for s ynchronization o f email and contact records 104 in the PDA 21 with email and contact records 247 (Figure 5).

The communication client 113 operates as a voice-over-internet-protocol VOIP compliant endpoint (VOIP endpoint) to enable the station 24 to provide real time communication services by establishing and maintaining VOIP media sessions with other VOIP endpoints. In the exemplary embodiment, the client 113 may be one of the commercially available clients utilizing established protocols such as the Internet Engineering Task Force (IETF) Session Initiation Protocols, or other protocols useful for signaling, establishing, maintaining, and tearing down VOIP media sessions utilizing UDP/IP channels o ver the IP compliant networks. More specifically, the client 113 may generate and respond to SIP compliant Invite, Ringing, OK, ACK, BYE, Cancel, and other SIP compliant messages known in the art.

In the exemplary embodiment, the communication client 113 may initiate a media session to a VOIP endpoint upon receiving identification of the VOIP endpoint (e.g. identification by extension number, subscriber name, or SIP URL

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identifier) via: i) subscriber activation of one of the selection buttons 60c (which corresponds to a predetermined identifier that is uniquely associated with the particular selection button 60c); ii) subscriber activation of the keypad 60a to input an identifier; iii) subscriber selection of an identifier from a menu on the display 59

5 using the touch panel 59b or menu and selection keys 60b; or iv) provision of the identifier from the system client 115.

The communication client 113 may further include DTMF tone generation module 111 and/or DTMF message module 121. The DTMF tone generation module 111 may generate a digital audio representation of a PSTN standard DTMF

10 tone in response to operator activation of one of the keypad buttons 60a. The digital audio representation may be coupled to the dialog system for output via the speaker 100 (or the speaker in the handset 130) to provide audio feedback of button 60a activation. The digital audio representation may also be coupled to the CODEC 122 wherein it may be compressed into frames for transmission to another

15 VOIP endpoint. Providing DTMF tones to another VOIP endpoint is useful for enabling a subscriber to place DTMF dial tones on one of the outside lines 33 and useful for enabling a subscriber to navigate through an audio prompt menu.

Because some DTMF tones may not be compressible (and subsequently reproducible) by widely used compression algorithms, the DTMF message module

20 121 may generate a message for transmission to another VOIP endpoint identifying a DTMF tone in response to subscriber activation of a keypad button 60a. Such message may be utilized by the other VOIP endpoint for generating the identified DTMF tone locally.

The status application 136 operates the indicators 139 to provide for each

25 indicator to properly display the "on-hook" or "off-hook" status of the resources (e.g. the real time communication device 15 or outside line 33 with which it is associated). The status application 136 also provides notice of the "on-hook" or "off-hook" status of the station to the control unit 12 and/or to other remote real time communication devices 15 that join a multicast group to which the status

30 application 136 announces the status of its local resources. Those real time communication devices 15 that join the multicast group utilize the status

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announcement to properly control their own local indicators 139. An exemplary status application 136 is shown in Figure 19 and described in more detail herein.

Wireless Telephony Device

5   Turning to Figure 3 in conjunction with Figure 1, exemplary structure of the wireless telephony device 26 is shown. The device 26 comprises a controller 50 coupled to a local bus 61 that interconnects the controller 50 with a plurality of peripheral circuits. The peripheral circuits may include a wireless module 94, a dialog system 53, a CODEC 122, a key switch controller 56, a touch panel

10   controller 58, a display controller 64, and a power supply 65. Further, coupled to the peripheral circuits, the device 26 may comprises a speaker 54 and a microphone 55 coupled to the dialog system 53 to provide a subscriber audio interface, buttons 57 configured as a telephone keypad and coupled to the key switch controller to provide subscriber key input, and a touch panel display 63

15   coupled to each of the display controller 64 and the touch panel controller 58 to provide a graphic subscriber interface.

The wireless module 94 may couple to the bus 61 either directly or through an interface circuit such as a PCMCIA controller, operate under control of applicable drivers operated by the controller 50, and, as discussed with reference

20   to the station 24, enable the device 26 to communicate with other devices over the network 22.

The CODEC 122 may couple to the bus 61, operate under control of applicable drivers and a packet voice client application 113 operated by the controller 50. As discussed with reference to the station 24, the CODEC 122

25   includes hardware circuits with adequate operating speed to: i) compress (and optionally encrypt) digital audio provided by the dialog system 53 into sequences of RTP frames for sending to another VOIP endpoint during a media session; and ii) sequence, decompress (and optionally decrypt) RTP frames provided by the other VOIP endpoint into digital audio for presentation to the dialog system 53.

30   The power supply 65 includes a battery and power supply circuitry. The power supply circuitry selectively receives input power from the battery and an

external power source and converts the input power to appropriate operating power required by each component of the device 26.

The dialog system 53 couples to the bus 61, operates under control of applicable drivers operated by the controller 50, and includes applicable circuits for: i) driving the speaker 54 in accordance with digital audio received from the compression encryption module 122, and ii) detecting input form the microphone 55 and generating digital audio for presentation to the compression encryption module 122.

The display controller 64 and the touch panel controller 58 each couple to the bus 61, operate under control of applicable drivers operated by the controller 50, and together enable the touch p anel display 6 3 to provide information to the subscriber and receive subscriber input. In the exemplary embodiment, the display controller 64 and the touch panel controllers 58 are each compatible with the display resolution and the touch panel resolution of the touch panel display 63,

The key switch controller 56 couples to the bus 61, operates under control of applicable drivers operated by the controller 50, and enables the controller 50 to receive subscriber input through the buttons 57.

The packet voice communication client 113 is operated by the controller 50 and, as discussed with reference to the station 24, operates to establish and maintain VOIP media sessions with other VOIP endpoints over the network 22.

The controller 50 may further operate the status application 136 (as shown in Figure 19) and may operate an indicator application 159. The indicator application 159 provides for virtual indicators 139 and virtual selection buttons 60c (as discussed with r espect to t he station 24) on the t ouch p anel d isplay 63. V irtual indicators and selection buttons may be utilized due to size constraints and power consumption constraints with indicators and selection buttons on a device the size of the wireless telephony device 26.

PDA

Referring to Figure 4, exemplary structure of the PDA 21 is shown. The PDA 21 may include all of the same structures as the wireless telephony device 26

discussed with reference to Figure 3, but may not include the packet voice communication client 113, the wireless module 94, the key switch controller 56, the buttons 57, or the dialog system 53, speaker 54, and microphone 55.

5	In addition to the elements discussed with reference to the telephony device 26, the PDA 21 include a wireless point-to-point communication module 62, a subscriber d evice a pplication 51, a n e mail and contact a pplication 6 6, and email and contact records 104 stored in a memory 103.

The wireless point-to-point communication module 62 couples to the bus 61 (either directly or through an interface circuit such as a serial communication 10	controller), operates under control of applicable drivers operated by the controller 50, and enables communication with a corresponding module 62 in a station 24.

The subscriber device application 51 is operated by the controller 50 and is similar to, and compatible with, the subscriber device application 119 of the station 24. The application 51 provides for synchronization (through the station 24) of 15	email and contact records 104 with email and contact records 247 in the control unit 12.

The email and contact a pplication 66 is o perated by t he controller 5 0 a nd provides for displaying information from the email and contact records 104 on the touch panel display 63 and for enabling subscriber manipulation of such records via 20	the touch panel display 63 or the buttons 57. The email and contact application 66 may be any commercially available email and contact client that is configured for operation on a small size display screen.

Control Unit

25	Referring to Figure 5 in conjunction with Figure 1 an exemplary control unit 12 is shown. The control unit 12 includes a service provider module interface 14 for coupling to the service provider interface 16, the PSTN interface 13, the local area network circuit 29, a packet audio/video gateway 232 (comprising a PSTN gateway 131, a session initiation proxy server 227, a service provider gateway 229, and a 30	master status application 219), a conference server 237, a voice mail server 236, an auto attendant module 239, an email module 228, a network address translation

server 334, an address server 220, a web server 230, storage 235, and communication services 31.

Some illustrative examples of a service provider interface 16 include: i) cable modem module 16a for communicating over coaxial cable 36 with a coaxial cable

5   based service provider network 18, ii) wireless radio module 16b for communicating over a wireless communication channel 38 with a service provider access point of a satellite or terrestrial wireless based service provider network 18; iii) a customer service unit (CSU) 16c for communication over a T1 line 40 with a digital PSTN based service provider network 18; and a fiber optic 16d for communication over a

10   fiber optic based service provider network 18.

The PSTN interface 13 couples to the one or more telephone lines 33 from the central office of the PSTN 42 and couples to the PSTN gateway 131. The PSTN interface 13 comprises applicable circuits for interfacing with the telephone lines 33 under control of the PSTN gateway 131 including, but not limited to, circuits

15   for: i) taking each telephone line 33 off hook to initiate a PSTN telephone call or to respond to PSTN ringing provided by the central office; ii) detecting dial tone on the telephone line 33 and providing a digital representation of the dial tone to the PSTN gateway 131; iii) modulating DTMF tones onto the telephone line 33 in accordance with a digital audio representation of the tones (or other applicable instructions)

20   provided by the PSTN gateway 131; iv) modulating audio (analog or PSTN digital audio) onto the telephone line 33 in accordance with a digital representation of audio provided by the PSTN gateway 132; and v) detecting modulated audio (analog or PSTN digital audio) on the telephone line and providing a digital representation thereof to the PSTN gateway 131.

25   Each of the PSTN gateway 131, the session initiation proxy server 227, the service provider gateway 229, the master status application 219, the conference server 237, the voice mail server 236, the auto attendant module 239, the email module 228, the network address translation server 234, the address server 220, and the web server 230 exchange information with each other and with remote

30   applications operating on remote devices coupled to the network 22 or the service provider network 18 utilizing TCP/IP connections and UDP/IP channels. As such,

communication s ervices 3 1 c omprise applicable I P stacks and port management systems for enabling such communication between components and provides for interfacing between the components and each of a network interface circuit 29 and the service provider module interface 14 for setting up TCP/IP connections and

5   UDP/IP channels over the network 22 and the service provider network 18 respectively.

The a ddress server 220 and the t ranslation server 234 e nable the control unit 12 to operate the network 22 as an IP subnet. The address server may be a known DHCP server that operates to assign IP addresses to the network devices

10   20. The translation server 234 may be a known IP layer proxy (e.g. NAT Server) enabling the various devices 20 to establish TCP/IP connections and UDP/IP channels to devices coupled to the service provider network 18.

A media session between two VOIP endpoints may be a VOIP session directly between the two VOIP endpoints. A media session between a VOIP

15   endpoint a nd a c ircuit switched device c oupled t o the P STN 4 2 may comprise a VOIP session between the VOIP endpoint and the PSTN gateway 131 plus a PSTN session between the PSTN interface 13 and the circuit switched device.

As such, the PSTN gateway 131 comprises: i) PSTN interface control circuits 132 which enable the PSTN gateway 131 to control the PSTN interface 13

20   to operate as a PSTN endpoint to a PSTN session with a remote circuit switched device over the PSTN 42, ii) a VOIP client circuit 135 that enables the PSTN gateway 131 to operate as a VOIP endpoint to a VOIP session with a corresponding VOIP client (either another device on network 22, the conference server 237, the voice mail server 236, or the auto attendant 239), and iii) a

25   translation circuit 133 (including a CODEC 122 and DTMF tone module 129) that translates audio between the PSTN session and the VOIP session.

The service provider gateway 229 operates as an IP layer proxy for translating frames between a VOIP endpoint on the network 22 and a VOIP endpoint on the service provider network 18 or the Internet 30.

30   The session initiation proxy server 227 facilitates set up of a VOIP session between two VOIP endpoints by routing session signaling messages there

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between. The session initiation proxy server 227 receives a session signaling message, refers to a routing table 245 for determining a routing address for the message based on an identifier within the message, translates the destination address within the message to the routing address, and forwards the message to

5   the routing address.

Turning briefly to figure 7, a flow chart representing exemplary operation of one aspect of the PSTN gateway 131 to couple a VOIP session to a PSTN session is shown.

Step 190 represents receiving call signaling over the network 22 from a real

10   time communication device 15 (e.g a signaling endpoint) that is signaling the PSTN gateway 131 on the PSTN Gateway 131 IP address and on a port number associated with one of the outside lines 33. Step 191 represents providing for the VOIP client 135 to establish logical channels to support a real time communication session with the signaling endpoint. Step 192 represents starting a PSTN session

15   by providing for the PSTN interface 13 to take the outside line 33 that is associated with the port on which the session signaling was received "off hook".

Step 193 represents providing a signal of the status of the outside line 33 to the status application 136 (Figure 19 discussed herein) of the master status application 219 such that it can send multicast status message 179 (Figure 23

20   discussed herein) announcing the state change of the outside line 33 resource to all real time communication devices 15 that have joined the multicast group associated with the outside line 33 resource.

Step 194 represents starting translation between the PSTN session and the VOIP session which may include detecting and receiving a dial tone on the outside

25   line 33 (as generated by the central office) and providing a digital representation thereof on the VOIP session (e.g. provides a digital representation of the dial tone to the CODEC 122 for compression into frames for transmission to the VOIP endpoint over the VOIP session). Translation may also include detecting dual tone multi frequency (DTMF) signals within the compressed audio data of the VOIP

30   session (or detecting messages representing DTMF signals from the endpoint) and providing for the DTMF tone module 129 to generate a digital audio representation

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of the DTMF tones for coupling to the PSTN interface 13 which in turn generates PSTN DTMF signals on the outside line 33 for dialing on the outside line 33 or menu navigation.

Step 195 represents determining whether the PSTN session is terminated by the central office (e.g. other PSTN endpoint hangs up) or whether it is terminated by the PSTN gateway 131 in response to termination of the VOIP session. When terminated, step 196 represents updating the status of the outside line 33 by providing a signal to the status application 136 of the master status application 219.

Turning to Figure 6, an exemplary routing table 245 is shown. The routing table 245 associates a routing address 254 to each resource 246 which represents a VOIP endpoint (e.g. real time communication device 15) or an outside line 33 of the system 10.

More specifically, the routing table 245 comprises a record 251 for each resource 246 that may be used to identify a real time communication device 15 or an outside line 33. The resource 246 may be a number such as a traditional extension number, a name, SIP URL, or other identifier of a person or line (collectively, a name). If a name and number correspond to the same destination, a single record 251 may be used.

Associated with each resource 246 is: i) a routing address 254; and ii) optionally a reference resource 248. The routing address 254 comprises an IP address 254a and a port number 254 that may be utilized by the session initiation proxy 227 to provide session signaling to the endpoint in response to receiving session signaling for the resource. The reference resource 248 is the resource to which the initiating endpoint is referred if the endpoint associated with the routing address 254 does not respond to the session signaling.

Referring to the flowchart of Figure 8 in conjunction with Figure 6, exemplary operation of the session initiation proxy server 227 is shown. Step 300 represents receiving session signaling, such as a SIP Invite message, from an initiating VOIP endpoint that includes an identification of a resource.

Step 302 represents determining the routing address 254 associated with the resource using the routing table 245 and step 304 represents translating the

destination address of the session signaling message to provide session signaling to the destination VOIP endpoint routing address 254.

Step 306 represents determining whether there is a response to the session signaling prior to elapse of a timeout duration. In the exemplary embodiment, a SIP 200 OK message may be considered a response, however, SIP ringing messages may not be considered a response.

If the destination endpoint at the routing address responds, step 308 represents translating the destination address in the response message to provide a response to the initiating endpoint such that further session set up messages may be transferred between the initiating endpoint and the destination endpoint.

If the destination endpoint does not respond within the time out period, step 310 represents canceling the session signaling and providing the reference resource 248 to the initiating endpoint such that the initiating endpoint may provide a session signaling message to the endpoint identified as the reference resource 248.

After a VOIP session is established, communication of audio (and video) data between the two endpoints comprises compressing digital audio data into a sequence of RTP frames, optionally encrypting the RTP frames, and sending the RTP frames to the other endpoint utilizing UDP/IP datagrams on the negotiated channels. At the other endpoint, the UDP/IP datagrams are received, sequenced, and the RTP frames are recovered, decrypted if applicable, and decompressed to yield the digital audio data.

Returning to Figure 5, to support a VOIP session between a VOIP endpoint on network 22 and a VOIP endpoint on the service provider network 18 or the Internet 30, the translation module 229 includes circuitry for operating as an IP layer proxy for relaying UDP/IP datagrams between the two endpoints.

Status Applications

Referring to Figure 24, the master status application 219 comprises a master multicast group table 187 and a status application 136.

The master multicast group table 187 comprises a plurality of records 224.

Each record 224 associates a resource, identified by its resource identifier 246 to a multicast group 189 associated with the resource.

In one embodiment of the present invention, a single multicast group will be used by a control unit 12 (e.g. all outside line resources) and all of the real time communication devices 15 (e.g. all extension resources) to announce resource status changes. In this first embodiment, only a single multicast group will be associated with all of the resources listed in the master multicast group table 187.

In a second embodiment of the present invention, each of the real time communication devices 15 will be assigned to one of a plurality of multicast groups and announce resource status changes only to those devices that have joined the assigned multicast group. The control unit 12 will join each of the plurality of multicast groups to which each outside line resource is assigned and announce resource status changes to the to the multicast group assigned to the outside line resource.

This second embodiment facilitates a single control unit 12 providing gateway services to distinct groups of real time communication devices such that it would appear that each distinct group of real time communication devices is operating on an independent telephone system independent from the other distinct groups.

In a third embodiment of the present invention, each of the control unit 12 and each of the real time communication devices 15 will host its own multicast group and announce resource status changes only to those devices that have joined the multicast group. As such, each resource will be assigned a multicast group 189 that is distinct from the multicast group 189 assigned to other resources.

Referring to Figure 19, a block diagram of the status application 136 is shown. The status application 136 exists within the master status application 219 of the control unit 12 and within each real time communication device 15. The status application 136: i) sends multicast messages that announce the "on-hook" state (e.g. not participating in a media session) and "off-hook" state (e.g. participating in a media session) of a local resource; ii) receives multicast messages announcing the "on-hook" state and "off-hook" state of a resource; and

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iii) controls the display status of the indicators 139 to assure that the status of each remote resource is properly displayed on the user interface.

The status application 136 comprises a multicast host 214, a multicast client 211, a resource status table 216, and an indicator control 137. Although the
5   multicast host 214 and the multicast client 211 are represented as two separate functional modules, it should be appreciated that the functions of each could readily be combined into a single module.

Figure 20 is a flow chart that represents operation of the multicast host 214 in each of the three embodiments of the present invention. Step 170 represents
10   the multicast host 214 obtaining the applicable multicast group address(es).

In the first embodiment, only a single multicast group address is utilized by the control unit 12 and by each real time communication device 15. As such, the multicast group address may be hard coded into applicable software or permanently, or semi-permanently, stored in a flash memory or register. Step 170
15   represents obtaining the multicast group address as stored.

Alternatively, if the multicast group address is not hard-coded or permanently or semi-permanently stored, then step 170 may represent the multicast host 214 querying the master multicast group table 187 (Figure 24) of the master status application 219 to obtain the multicast IP address defining the single multicast
20   group associated with all of the resources 246 in the master multicast group tables 187.

In each of the second and third embodiments, multiple multicast group addresses are utilized. Therefore, step 170 would represent the multicast host 214 querying the master multicast group table 187 (Figure 24) of the master status
25   application 219 to obtain the multicast IP address (or addresses) defining the multicast group(s) to which the resource 246 local to the control unit 12 or the resource 246 local to the real time communication device 15 is assigned.

As discussed, in the second embodiment of the present invention, each of the real time communication devices 15 will be assigned to one of a plurality of
30   multicast groups and therefore will receive only one multicast group address at step 170. However, the control unit 12 will join each of the plurality of multicast groups

and therefore will receive a multicast group address for each of its outside line resources at step 170.

Each multicast group address will be within the block of multicast addresses that will remain local to the local area network 22, but could be forwarded by local
5   switches and routers within the local area network 22.   As such, the multicast address will be an address known as a limited scope address within the range of 239.0.0.0 through 239.255.255.255. The NAT server 234 of the control unit 12 is equipped with filters that prevent routing of multicast messages to the service provider network 18.

10   Step 171 represents the multicast host 214 entering the multicast group(s) assigned to each of its local resources defined by the multicast address(es). More specifically, a t step 1 71a the m ulticast h ost 214 must perform known n etworking operations to bind a port and assure that any routers or switches comprising the local area network 22 route multicast messages (for each multicast group joined) to
15   the particular subscriber station 24 or wireless telephony device 26. Step 171b represents sending a status refresh request to the newly joined multicast group(s).

After performing the s teps associated with entering the m ulticast group(s), the multicast h ost 214 enters an operational state represented by steps 172 and 174. Within the operational state, three events may cause the multicast host 214 to
20   send a multicast status message at step 174 to its assigned multi-cast group(s).

The first event, represent by step 172a, is receipt of a status refresh request. It should be appreciated that when each device 15 or the control unit 12 joins a multicast group, it sends a status refresh request at step 171. The purpose of the status refresh request is to initially obtain the status of every device within the
25   multicast group. Therefore, when a status refresh request is received at step 172a, a multicast status message 179 (Figure 23) is sent to the IP multicast address 189 on the network 23.

The second event, represented by step 172b, is a state change of a local resource. The multicast host 214 monitors those resources 246 that are local (e.g.
30   local resources) the local resources to detect a change in "on-hook" and "off-hook" state. When a state change is detected at step 172b, a multicast status message

179 (Figure 23) is sent to the IP multicast address 189 on the network 23.

The third event, represented by step 172c, is an elapse of time during which a multicast status message 179 has not been sent. The multicast host 214 may determine, at step 172c, that a time period has elapsed during which a multicast

5   status message has not been sent at step 173. In response to detecting that the time period has elapsed, the multicast status message may be sent at step 174. The process of step 172c provides for the multicast host 214 to send a message every fixed, or variable, period of time even if no state change has occurred. As such, a device that is newly coupled to the network 23 (that did not send a refresh

10   request message), or a device that did not receive a previous multicast status message 179, may receive a multicast status message 179 sent within a brief period of time to assure that its indicators 139 (associated with the resource) will properly reflect the state of the resource.

It should be appreciated that although Figure 22 is structured such that any

15   of the three events of step 172 would initiate the host sending a multicast status message, it is within design choice to select any one, two, or all three of the events (or similar events) in implementing the present invention.

Referring briefly to Figure 23, the multicast status message 179 announces the state of the resource 246. The message 179 includes various IP headers to

20   transport the frame to all other devices that have joined the multicast group, such headers including the IP multicast address 189 as a destination address. In addition, the multicast status message 179 may include the resource ID 246 and an indication of the state 182 of the resource 246.

Figure 21 is a flow chart that represents operation of the multicast client

25   211. Step 175 represents the multicast client 211 obtaining the applicable multicast group address(es) in a similar manner as the multicast host 214 obtained the multicast group address(es) as discussed with reference to step 170 of Figure 20.

Step 176 represents joining the multicast group(s) associated with each of its local resources in a similar manner as the multicast host 214 joined the multicast

30   groups as discussed with reference to steps 171a of Figure 20.

Thereafter, the resource client 211 enters an operational state depicted by

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steps 177 and 178. During the operational state, the multicast client 211 may receive multicast status messages 179. In response to receiving a multicast status message 179, the multicast client 211 will update its local resource status table 216 at step 178.

5   It should be appreciated that at start up, it is one possible implementation that the multicast host 214 sends a status refresh message on the multicast group address at step 171b, receives its own status refresh message at step 172a, and sends the status of its own local resources at step 174. As such, the resource client 211 obtains status of its own local resources at start up.

10   Referring to Figure 22, the resource status table 216 comprises a plurality of records 206, each of which is associated with one of the resources 246. In association with each resource 246 are: i) a status field 184 containing an "on-hook" or "off-hook" status of the resource 246, ii) an indicator field 185 idnetifying which of the indicators 139 are associated with the resource 246; and iii)

15   (optionally) a multicast group field 186 identifying the multicast group 189 associated with the resource.

In the first embodiment wherein the control unit 12 and all of the real time communication devices 15 utilize the same multicast group 189 for sending multicast status messages, the IP multicast address 189 in the multicast group field

20   186 for all resources 246 will be the same IP multicast address.

In the second embodiment, all local resources of, and all resources accessible to, a real time communication device will be assigned to a single distinct multicast group. As such, the real time communication device 1 5 will utilize the same multicast group 189 for sending multicast status messages. However, the

25   control unit 12 may have resources assigned to a plurality of the distinct multicast groups. As such, the control unit 12 may use multiple multicast addresses for sending status messages.

In the third embodiment wherein a distinct multicast group is established for each resource 246, then the IP multicast address 189 in the multicast group field

30   186 will be distinct for each resource 246.

Turning to figure 27, exemplary operation of the indicator control module 137

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is shown. The indicator control module 137 monitors the resource status table 216 to detect a change in the state of a resource 246. Upon detecting a state change at step 410, the indicator control module 137 updates the display of the indicators 139 that are associated with the resource 246 in the resource status table 216 to reflect

5 the state change.

Conference Server

A conference session comprises a real time communication session amongst participants such that each participant receives audio representing the

10 other participants. And, each participant with video display capabilities may receive video from other participants that have video capture capability.

Referring to Figure 9, to provide such conference session capabilities, the conference server 237 comprises a session client module 240, an audio mixer 238, a video control module 242, and a conference server application 244.

15 The session client module 240 operates as an endpoint to a VOIP session with each conference session participant. More specifically, the session client module establishes a session with each conference participant in accordance with an identification corresponding to each participant as provided by the conference server application 244 and then maintains each VOIP session for the duration of

20 the conference session.

The audio mixer 238 receives an audio stream from the session client module 240 for each conference participant and generates one or more conference mix signals. The conference mix signals are provided to the session client module for transmission to the conference participants.

25 The video control module 242 receives a video stream from the session client module 240 for each conference participant that provides motion video, generates a motion video stream for each participant in accordance with the participant's video selection (as received from the conference server application 244) and provides each such video stream to the session client module 240 for

30 transmission to the applicable conference participant.

The conference server application 244 controls operation of the audio mixer

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238, the session client module 240, and the video control module 242. The conference server application 244 also operates as a web server to provide a user interface to session participants that enable session participants to set up and control a conference session.

5   Figure 10 shows a flowchart that represents exemplary operation of the conference server application 244. Step 256 represents receiving a conference set up request from a subscriber at an initiating subscriber station 24 or a subscriber device 26 (initiating real time communication device 15 of Figure 1). In the exemplary embodiment, the conference set up request may take the form of the

10   initiating station sending a frame to the conference server application 244, on a predetermined IP address and port number, to establish a TCP/IP connection with the conference server application 244. Step 257 represents establishing the session with the initiating real time communication device 15.

        Step 258 represents retrieving the address book content 249 of the

15   subscriber associated with the initiating real time communication device 15 from the email and contact records 247 in storage 235. Step 260 represents providing a conference initiation document that includes the address book content 249 to the initiating real time communication device 15 in document format. The document format may be an HTML document, an XML document (e.g. content messages and

20   display layout control messages) or other document format displayable on the initiating real time communication device 15.

        Turning briefly to Figure 11a, an exemplary conference initiation document 287 is shown. The document 287 includes a record 288 for each contact (or contact group) from the subscriber's address book content and a set up call control

25   289. A scroll control 290 enables display of additional records 288 that may not fit entirely on the display of the initiating real time communication device 15. The document 287 also includes applicable script to enable the subscriber to highlight multiple records 288 to select multiple conference session participants using the touch panel 59b or the navigation and selection buttons 60b. And, applicable script

30   such that when the subscriber activates the set up call control 289, the selected records 288 are identified to the conference server application 244 over the TCP/IP

connection.

Returning to Figure 10, step 262 represents receiving the participant list and step 264 represents setting up a session status table for the conference session. Turning to Figure 13, an exemplary session status table 291 is shown. The session status table 291 includes a record 292 for each conference session participant. Associated with each participant is the participant's name 293 and identifier 294 for setting up a media session to the participant, both from the email and contact records 247. Further associated with each participant is an indication of each of the participant's security status 295, the participant's audio status 296, the participant's video status 297, and the participant's video display mix selection 298, each of which is discussed in more detail herein.

Step 266 represents providing the identifier 294 for each participant from the session status table 291 to the session client module such that the session client module may initiate a media session to each participant.

Step 268 represents building a session status document for each participant. Turning to Figure 11b, an exemplary session status document 311 is shown. The session status document 311 includes a record 312 for each participant and associated with each participant is an indication of the participant's status. The participants status may be: "A" indicating that the participant is active in the session sending and receiving audio; "I" indicating that the participant is inactive (e.g. no session established with the participant); "M" indicating that the participant is active, receiving an audio stream, but is muted and not sending an audio stream; and "P" indicating that the participant is in a separate and private break-out conference session with one or more other participants.

Step 270 represents inviting each participate to access its status document 311. More specifically, the conference server application 244 may make each participant's status document 311 available on a predetermined port number and may provide a frame to each participant inviting that participant to establish a TCP/IP connection at the particular port number where such participant's status document 311 is available

Steps 272, 278, and 280 together represent the conference server

application 244 waiting for a video request event, an encryption request event, and a session status change event.

A video request event corresponds to a subscriber selecting video display parameters and activating the video control 314 on the participant's session status document 311. More specifically, the subscriber may utilize records 312 (and scroll control 315) to highlight multiple (up to four) records associated with participants that are providing video and then activate the video control 314. Scripting in the session status document 311 will provide the video request (that includes identification of the participants associated with the highlighted records) to the conference server application 244 in response to activation of video control 314.

An encryption request event corresponds to a subscriber activating the encryption control 319 on the session status document 311. In response to such activation, script in the session status document 311 will provide the encryption request to the conference server application 244.

A session status change event may be any of a participant entering a session (starting a VOIP session with the session client module 240), departing from a session (terminating a VOIP session with the session client module 240), muting participation in a session (continuing the session but providing nil audio and/or video), beginning or terminating the provision of motion video to the session client module 240, or beginning or terminating of encryption of a session with the session client module 240. Each session status change event may be reported to the conference server application 244 by the session client module 240 and will include an indication of the change event.

In response to a video request event, at step 274 the conference server application 244 will build a video display document and provide the video display document to the participant.

Figure 12a represents a first video display document 150a that includes a video frame 151, a return to status document control 152, an embedded port number 153 that corresponds to the port number on which the session client module 240 has made the full motion video provided by the participant identified in the video request event available, and embedded scripting 154 that provides for the

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participant to connect to the port number 153 and display the motion video provided on the port within the frame 151. The return to status document control 152 includes embedded scripting such that upon activation, a request is sent to the conference server application 244 to obtain the session status document 311 as
5   shown in Figure 11b.

Figure 12b represents a second video display document 150b that includes four video frames 155a - 155d, a return to status document control 152, four embedded port numbers 156 that corresponds to the port numbers on which the session client module 240 has made the full motion video provided by the four
10   participants identified in the video request event available, and embedded script 157 that provides for the participant to connect to each of the port numbers 156 and display the video provided on each port in one of the four frames 155a - 155d. The return to status document control 152 includes embedded script such that upon activation, a request is sent to the conference server application 244 to again
15   obtain the session status document 311 as shown in Figure 11b.

Returning to Figure 10, in response to an encryption event at step 278, the conference server application 244 will provide a signal to the session client module 240 to initiate encryption with each participant at step 282.

In response to a session status change at step 280, the conference server
20   application 244 will update the status table 291 (Figure 13) and each status display to correspond to the changed status at step 284 and will provided the updated session status document 311 to each participant.

Web Server
25   The web server application 230 provides multi media communication services to each subscriber which may include: a) updating of the network location table 245 to assure proper routing of incoming audio and audio/video calls; b) proxy communication over network 18; c) delivery of a multicast messages directed to a subscriber to the particular station 24 at which his or her subscriber device is then
30   currently coupled; and d) providing a menu control for access to the conference server application 244 and the email module 228.

To perform such functions, the web server 230 includes a management application 226 and a multicast application 231. The flowcharts of Figures 14a through 14c represent exemplary operation of the management application 226.

Referring to Figure 14a, step 320 represents the web server 230 receiving an open session request from a telephony station 24 or a wireless telephony device 26 (device 24, 26) that has been operatively coupled to network 22 and is ready to operate as a client of the web server 230. The open session request may take the form of a frame sent by the device 24, 26 to a predetermined port number to open a TCP/IP session with the web server 230.

Step 322 represents retrieving a main menu document from document storage 241. The main menu document may be an HTML document or an XML document (e.g. content messages and display layout control messages) or other document format displayable on the device 24, 26. Step 324 represents providing the main menu document to the device 24, 26.

Figure 15 represents an exemplary main menu document 160. The main menu document 160 includes a plurality of subscriber activated controls 162 that may include an email control 162a, a voice mail control 162b, a multicast paging control 162c, a conference call control 162d, and an address book control 162e.

The email control 162a, the voice mail control 162b, and the address book control 162e each include script 163a, script 163b, and 163e respectively such that when activated, a message is provided to the email module 228 on a predetermined port number indicating such activation. The multicast paging control 162c may include script 163c such that when activated, a message is provided to the multicast application 231 on a predetermined port number indicating such activation. The conference call control 162d may include script 163d such that when activated, a message is provided to the conference server application 244 indicating such activation.

Referring to Figure 14b, step 326 represents the management application 226 receiving an indication that a PDA 21 has coupled to a station 24 via the point-to-point communication module 62. The indication may take the form of a frame sent by the subscriber station 24 to a predetermined port number.

Step 328 represents the management application 226 providing script to the station 24 that provides for the station 24 to obtain the PDA device ID number associated with the PDA 21 and report the PDA ID number back to the management application 226. Step 330 represents receiving the PDA ID number.

5   Step 332 represents associating the PDA 21 with the station 24 in the routing table 245 by adding the PDA ID number to the record associated with the station 24.

Step 334 represents providing instructions to the station 24 to activate the subscriber device application 119 on the station 24 to initiate a link with the

10   synchronization application 51 on the PDA 21 and step 336 represents synchronizing email records and contact records 104 in the PDA 21 with the email and contact records 247 in the control unit 12.

Referring to Figure 14c, step 338 represents the management application 226 receiving an indication that a PDA 21 has de-coupled from a station 24. The

15   indication may take the form of a frame sent by the station 24.

Step 340 represents dissociating the PDA 21 from the station 24 in the routing table 245.

The flowchart of Figure 17 represents steps performed by the multicast application 231 upon receiving a message indicating subscriber activation of the

20   multicast control 162c. Step 374 represents receipt of such a message.

Step 376 represents retrieving the subscriber's address book content 249 from the email and contact records 247 and step 378 represents retrieving a select paging group document from document storage 241. Step 380 represents providing the select paging group document including the subscriber's address

25   book content 249 to the station 24.

Figure 18 represents an exemplary select paging group document 75. The document 75 includes a plurality of records 76 (that embody the address book content 249), a scroll control 77, a main menu return control 78, and a start message control 79. E ach record 76 is associated with a paging group. Some

30   paging groups may include only a single name such that individuals may be selected to include in the multicast page and some paging groups may include

multiple individuals (or multiple sub groups). The document further includes embedded scripting 80 which enables the subscriber to select (using a highlight bar control and the scroll control 77) one or more groups for a multicast paging message. The start message control 79, includes embedded script that provides

5   for identification of the selected paging groups to be provided to the multicast application 231 upon activation.

Returning to Figure 17, step 382 represents receiving the identification of the selected paging groups. Step 384 represents obtaining a routing address for each selected group participant from the routing table 245 and step 386 represents

10   sending a message to invite each group participant to the multicast session group using the routing address determined for each group participant at step 384. Step 386 represents receiving response messages from the group participants.

Step 388 represents set up of an RTP channels to each response address received from a group participant at step 386.

15   Step 390 represents prompting the initiating subscriber station 24 to begin the message and step 492 represents providing the message in multicast format on each RTP channel.

Voice Mail

20   Returning to Figure 5, the voice mail module 236 includes VOIP client circuits for responding to the call signaling provided by the call signaling module 227, maintaining a VOIP session with an initiating VOIP endpoint, providing a sequence of RTP frames representing applicable audio prompts from compressed audio prompt files 233 to provide a typical voice mail experience to the operator of

25   the initiating endpoint, to receive RTP frames from the initiating VOIP endpoint representing the remote caller leaving a message for the subscriber, and to compress the message into a digital audio file for storage. The voice mail message contained in the digital audio file can be retrieved in a traditional manner by calling into the voice mail server. Alternatively, the voice mail module 226 may send the

30   digital audio file to the email module 228 for storage in the inbox 250 for later retrieval by the subscriber.

E-Mail

The email module 228 maintains an email account associated with each subscriber. The email module 228 includes client circuits for interfacing with a

5   remote email server for receiving email messages for each subscriber and for storing in the subscriber inbox, and sends email messages drafted by the subscriber. The email module 228 also maintains the email files 247 in the storage 235 that may include the address book content 249 and the inbox 250 for each subscriber.

10   The email module 228 may be any commercially available email server that supports inbox and address book functionality and provides client services through a web document interface. Because the main menu document 160 includes separate controls for voice mail, email, and address book, the flow charts of Figures 16a and 16b represents exemplary operation of the email module 228 upon receipt

15   of a message indicating one of such controls from the main menu document 160.

The flowchart of Figure 16a represents steps performed by the email module 228 upon subscriber activation of the email control 162a or the voice mail control 162b. Step 356 represents receipt of a message indicating such activation from a subscriber station 24.

20   Step 358 represents a determination whether the identity of the subscriber at the subscriber station 24 is recorded in the routing table 245. If a PDA 21 is associated with the station 24 or if a subscriber is associated with the station 24 in the routing table 245, the system proceeds directly to step 364 wherein inbox content 250 associated with the subscriber is retrieved from the email and contact

25   records 247. However, if a subscriber is not associated with the station 24 in the routing table 245, the system proceeds to step 260 wherein a logon screen is provided to the station 24 and a subscriber identifier is received from the station 24 at step 362. Then the system proceeds to step 364.

Step 366 represents a determination whether the subscriber activated the

30   email control 162a or the voice mail control 162b. If the subscriber activated the email control 162a, the inbox content 250 is sorted such that email messages

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(messages other than those with voice mail attachments received from the voice mail server 236) are displayed at the top of the list at step 370. Alternatively, if the subscriber activated the voice mail control 162b, the inbox content is sorted such that the email messages that include voice mail attachments are displayed at the top of the list at step 368.

Step 372 represents retrieving a messaging document template from document storage 241 and providing a messaging document with the sorted inbox content 250 embedded with the messaging document template to the station 24.

The flowchart of Figure 16b represents exemplary steps performed by the email module 228 upon receipt of a message, at step 342, indicating subscriber activation of the address book control 162e.

Step 344 represents a determination whether the identity of the subscriber at the subscriber station 24 is recorded in the routing table 245. If a PDA 21 is associated with the station 24 or if a subscriber is associated with the station 24 in the routing table 245, the system proceed directly to step 350 wherein address book content 249 associated with the subscriber is retrieved from the email and contact records 247. However, if a subscriber is not associated with the station 24 in the routing table 245, the system proceeds to step 346 wherein a logon screen is provided to the station 24 and a subscriber identifier is received from the station 24 at step 348. Then the system proceeds to step 350.

Step 352 represents retrieving an address book document template from document storage 241 and providing the address book content 249 embedded in the address book document template to the station 24.

Conclusion

It should be appreciated that the systems and methods of the present invention provides a simulated key switch private telephone system user experience in a VOIP environment. Although the invention has been shown and described with respect to certain preferred embodiments, it is obvious that equivalents and modifications will occur to others skilled in the art upon the reading and understanding of the specification. It is envisioned that after reading and

understanding the present invention those skilled in the art may envision other processing states, events, and processing steps to further the objectives of the modular multi-media communication management system of the present invention. The present invention includes all such equivalents and modifications, and is

5   limited only by the scope of the following claims.